VoIP GSM Gateway

User Manual

Altesys

【Content】

1. Introduction

GSM GATEWAY series products provide you the best connect solution for heterogeneous network(including : WLAN、GSM、CDMA or PSTN)You may use a SIP-protocol VoIP phone or software to connect to the GSM GATEWAY, then reach this call to the mobile network, and vice versa. With multiple sets of GSM GATEWAY, you may even build an international call network.

2. Function description

2.1 VoIP(SIP) to GSM conversion.

2.2 50 sets of LAN->MOBILE routes setting，
50 sets of MOBILE->LAN routes setting.

2.3 Voice response for setting and status (dial in from mobile).

2.4 Series connections to save bills.

2.5 Standard SIP(RFC2543,RFC3261) protocol，communicates with other gateway or PC.

3. Parts list

Please check the parts for any missing parts. If do, please contact our agents：

3.1 「GSM GATEWAY」main body

3.2 Power adaptor AC-DC (110V AC – 12V DC) or (220V AC – 12V DC)

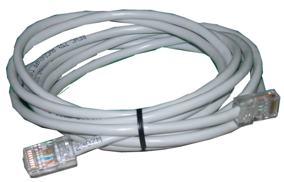
3.3 Network cable

3.4 Antenna

3.5 User Manual

3.1

3.2



3.3

3.4

4. Dimension

5. Chart of the device

(5.1) (5.2) (5.3) (5.4) (5.5) (5.6) (5.7)

5.1 Antenna : Antenna connector.

5.2 DC 12V : Power input.

5.3 LAN : RJ-45 internet connector , standard RJ-45 socket , connect to HUB.

5.4 PWR (Power LED) : Light up when power is normal.

5.5 MOBILE Indicator : Normally it represents the signal strength. It flashes 5 times per second for the strongest signal, and once per second for the weakest signal. It lights up during the startup period. When the call is coming from the MOBILE, it flashes on and off for 0.5 second.

5.6 LAN Indicator : Off in normal time. When the call is coming from the

LAN, it flashes on and off for 0.5 second.

5.7 LINK Indicator : Light up when network is connected.

6. CABLING

6.1 Connect the internet cable from HUB/Switch to the 'LAN' connector of the GSM GATEWAY.

6.2 Connect the antenna and put it in proper position to get the best signal reception.

6.3 Insert the SIM card from back of the main body.

6.4 Connect the power adaptor. The 'POWER' LED should be light up.

7. IP Setting

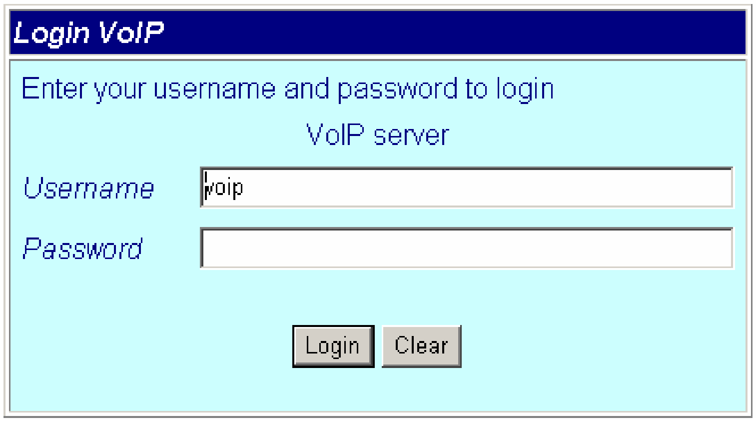
The operator can setup or query the network parameters by dialing in the mobile number which it SIM card has been put in the main body. The status or result is response by voice. In the first 20 seconds after power-on, the GSM GATEWAY enters the IP setting mode. The operator may dial in the mobile number during this period to set or query the network parameters.

Item	Function	Code	Remark
1	Password Check	**#000+[number]#**	Need to pass the password check in order to do the rest functions. The default password is "1234"
2	Reboot	**#195#**	Reboot the GSM GATEWAY
3	Factory Reset	**#198#**	Resume to original factory setting
4	Check IP Address	**#120#**	Check the IP address, Default : 192.168.0.100
5	Check IP Type	**#121#**	Check the DHCP On/Off flag, default : OFF
6	Check Network Mask	**#123#**	Check the MASK, Default : 255.255.255.0
7	Check Gateway IP Address	**#124#**	Check the Gateway IP address, Default : 192.168.0.254
8	Check Primary DNS Server	**#125#**	Query Primary DNS Default : 192.168.0.1
9	Check Firmware Version	**#128#**	Query the firmware version number
10	Set as DHCP client	**#111#**	Set as DHCP client
11	Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Set IP address(3 digits for each field, prefix by 0 if not sufficient)
12	Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Set the MASK(3 digits for each field)
13	Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Set the Gateway IP address (3 digits for each field)
14	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Set the Primary DNS (3digits for each field)

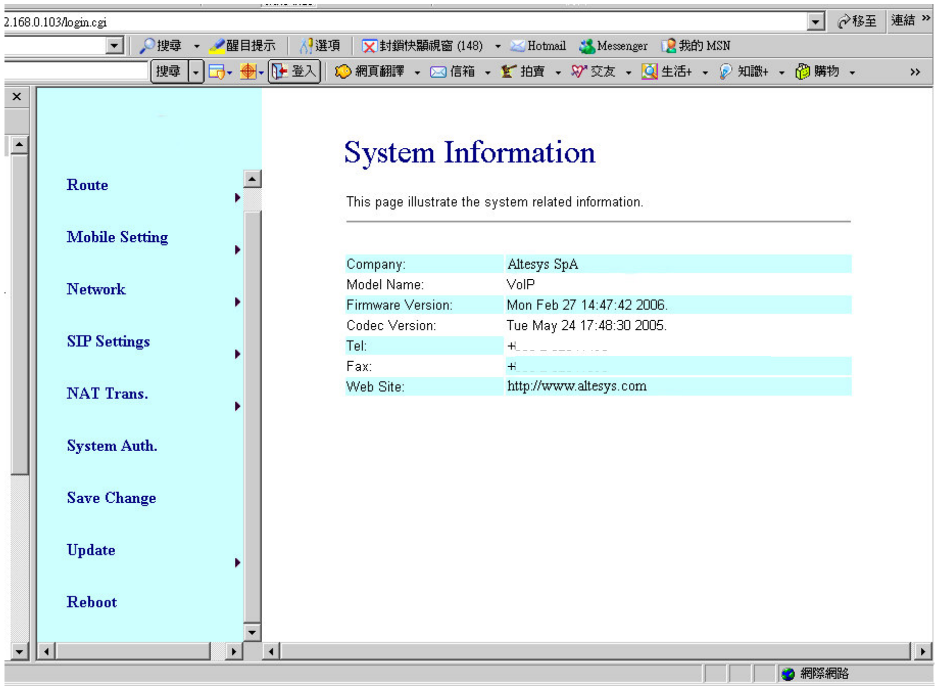
8. Web Page Setting

When the IP setting is done, the operator may setup all the rest parameters via web page. Browse the IP address from Internet Explorer (default is http://192.168.0.100).

The following page shows up :



Enter the username and password for authentication (default username=**voip**, password=**1234**).The page follows when the username and password are correct.



9. Network

In Network you can check the Network status, configure the Network Settings and DDNS settings.

Network Status: you can check the current Network setting in this page.

Network Settings: You can configure the IP Phone Network setting in this page.

The **TCP/IP** Configuration item is to setup the LAN port's network environment. You may refer to your current network environment to configure the IP Phone properly.

The **PPPoE** Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

The **Bridge** Item is to setup the IP Phone Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.
When you finished the setting, please click the Submit button.

DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.

10. SIP Setting

In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related information correctly then you can register to the SIP Proxy Server correctly.

In **Service Domain** Function you need to input the account and the related information in this page, please refer to your ISP provider. You can register three SIP account in the GSM Gateway. You can dial via first enable SIP account and receive calls from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

Display Name: you can input the name you want to display.

User Name: you need to input the User Name get from your ISP.

Register Name: you need to input the Register Name get from your ISP.

Register Password: you need to input the Register Password get from your ISP.

Domain Server: you need to input the Domain Server get from your ISP.

Proxy Server: you need to input the Proxy Server get from your ISP.

Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

You can see the Register Status in the Status item. If the item shows "Registered", then your IP Phone is registered to the ISP, you can make a phone call directly.

If you have more than one SIP account, you can following the steps to register to the other ISP.

When you finished the setting, please click the Submit button.

Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

Codec ID Settings: you can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.

DTMF Setting: you can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

RPort Function: you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

11. Mobile Status

Network Registration: identifies the telecom carrier which the SIM card been registered.

SIM Card ID: SIM card ID.

Signal Quality: Signal quality.

LAC CI.: Local Area Code of the base station and the Channel ID.

Income IP: The IP address of the last incoming call from LAN.

Outgoing IP: The IP address of the last outgoing call to LAN.

Income TEL: The caller ID of the last incoming call from MOBILE.

Outgoing TEL: The called number of the last outgoing call to MOBILE.

12. Mobile Setting

VoIP Handset Volume: VoIP volume adjustment (0 –15)

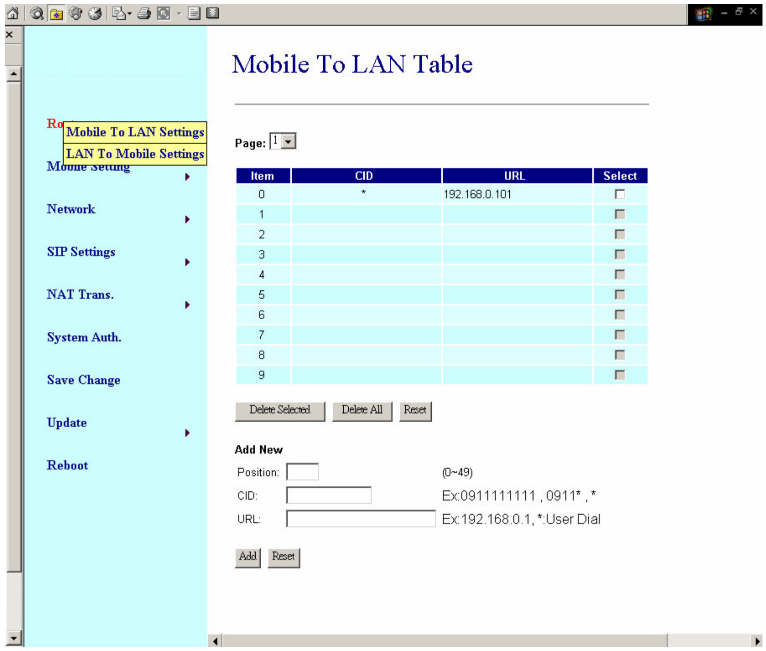
VoIP Handset Gain: VoIP gain adjustment (0 –15)

Caller ID: You may select to display the Caller ID from GSM incoming call, or fixed SIP user name.

Mobile PIN Code: If you need to unlock pin code via GSM GATEWAY,you can click "On" and enter pin code.

13. Route Mobile->LAN Settings

The operator may assign 50 sets of routing rule to transfer the call incoming from MOBILE to LAN.



The GSM GATEWAY will transfer to the URL according to the caller ID of the Mobile.

***CID:** may enter the whole number (e.g. 0911111111) or, only part of the number (prefix) e.g. 0911* means any number starting with 0911 will be accepted

or, * means all numbers can be accepted

or, N means the calls without the CID

Please note the priority of the rules. The item which has more digits will have higher priority. If the digits are the same, then former one gets the higher priority.

***URL:** The IP address or the extension number to transfer this call.

If the GSM Gateway is registered to a Proxy Server (VoIP PBX) this field should contain the internal extension number or queue number (es: 100) you want the gateway to transfer the call to.

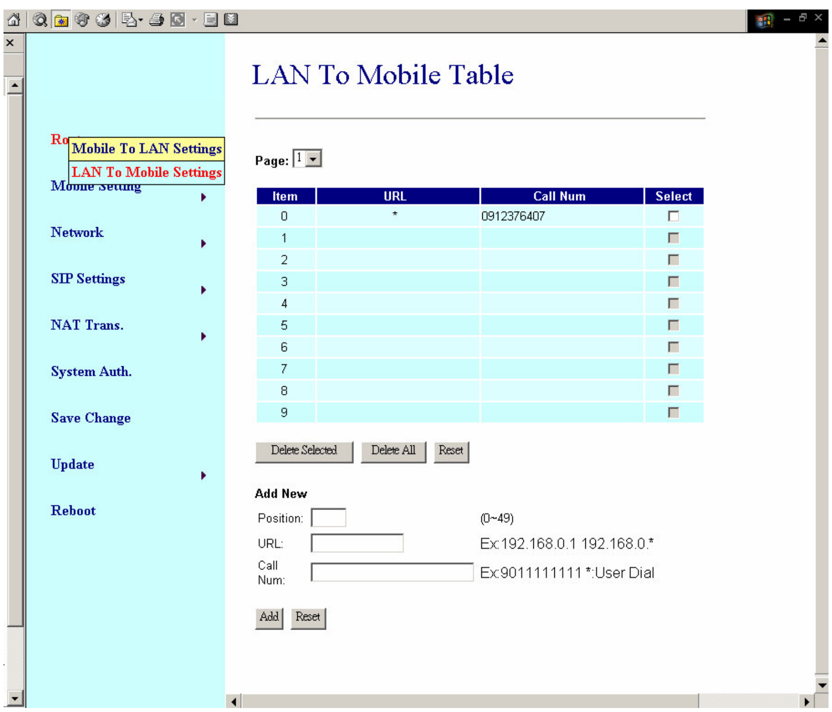
If the GSM Gateway is not registered to a Proxy Server (VoIP PBX), this field should refer to one of the Speed Dial Number configured in "Mobile to LAN Speed Dial" Menu (a number from 0 to 9).

If this field is blank or simply 'N', it means refuse to transfer.

If an '*' entered, it means 2-stages-dialing. The call will be answered and the Gateway will prompt a dial tone again to receive the number to call (es: 024888631 or 100): the call will be forwarded as from an IP-Phone.

14. Route LAN ->MOBILE Settings

The operator may assign 50 sets of routing rule to transfer the call incoming from LAN to MOBILE.



The GSM GATEWAY will transfer to the mobile number according to the incoming URL

***URL** : The IP address or the extension number of the incoming call (can be retrieved from “Mobile”>”Status” soon after a call).

May enter the whole IP address, e.g. 192.168.0.101 or user . If a simple ‘*’ is entered, means no restriction for the incoming IP address.

***Call Num** :

1. may enter the whole number, e.g. 0911111111
2. a simple *”means 2-stages-dialing. The call will be answered and prompt dial tone again to receive the called number as the destination, e.g. 0911111111 or 0911111111#
3. #['d'n]['a'ppp] for one-stage (direct) dialing
   [...] means this is an optional rules.
   'd'n means to delete the beginning n codes,
   'a'ppp means to add 'ppp' in front.
   for example #d2a09 means one-stage dialing,
   delete the first 2 codes from your destination number,
   then add 09 in front as the new destination number.

15.Specification

15.1 Protocols

SIP (RFC2543,RFC3261)

15.2 TCP/IP

IP/TCP/UDP/RTP/RTCP/
CMP/ARP/RARP/SNTP
DHCP/DNS Client
IEEE802.1P/Q

ToS/DiffServ

NAT Traversal

STUN

uPnP

IP Assignment

Static IP

DHCP

PPPoE

15.3 Codec

G.711 u-Law

G.711 a-Law

G.723.1 (5.3k)

G.723.1 (6.3k)

G.729A

G.729A/B

15.4 Voice Quality

VAD

CNG

AEC, LEC

Packet loss

15.5 GSM

Dual Band: EGSM 900/DCS 1800 MHZ

Tri Band: EGSM 900 and GSM 1800(GSM Phase 2+) and DCS 1900

Speech Service with EFR (Enhance Full Rate)/FR (Full Rate)/HR (Half Rate) Codec.

16.Setup GSM GATEWAY with Asterisk

16.1 Usage

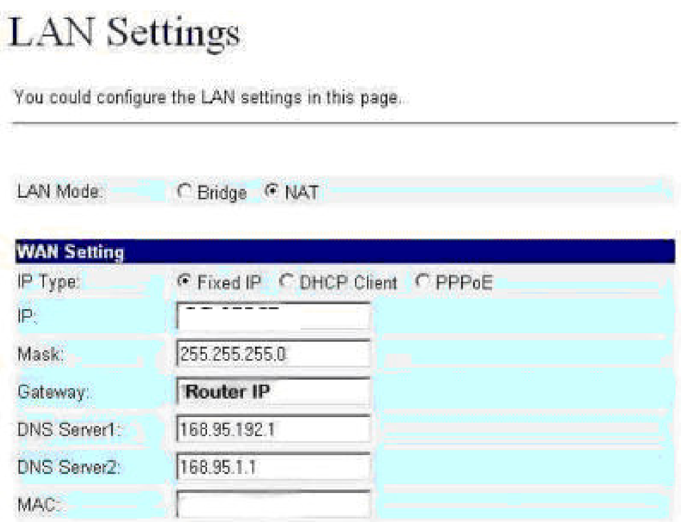
Apart from outgoing calls, a typical usage of such a gateway is to be able to give a call with your normal mobile to any destination at voip cost :
Your mobile <----*gsm network*----> GSM GATEWAY <--*lan*--> Asterisk <--*internet*--> VOIP provider <--*whatever*--> landline

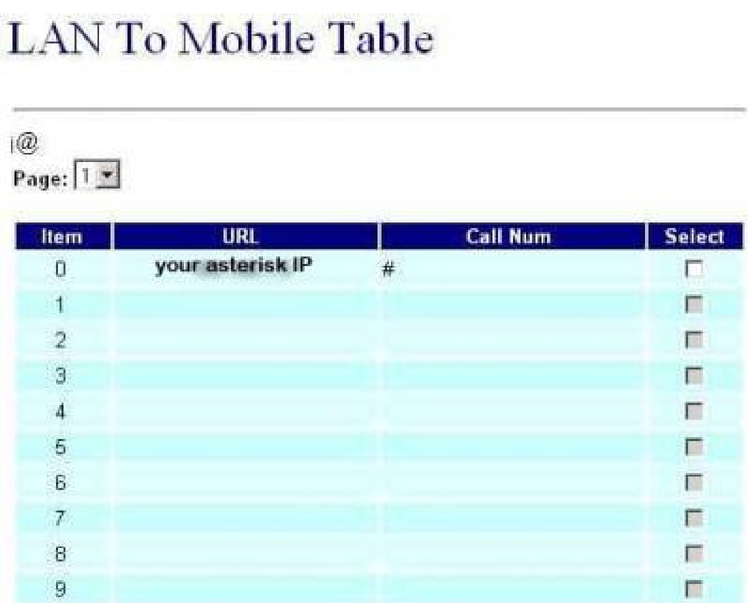
To do such a call, you just call your GSM GATEWAY number (it has its own simcard), then you get an invitation tone, then you dial the number which is handled by Asterisk.
If you have some special deals with your mobile operator, like free special number, you can call your GSM GATEWAY for free.

16.2 GSM GATEWAY Configuration

Once you've configured everything in the box, one good advice is to restart it. By this way you should have all the parameters taken into account.

To have the GSM GATEWAY to work with Asterisk, you need first to configure the box.
Here are some screen shots showing all the important parameters.
You have to note that in all the configuration process, the GSM GATEWAY is considered as extension '103' of the IPBX.
In **Bold** are the parameters depending on your installation





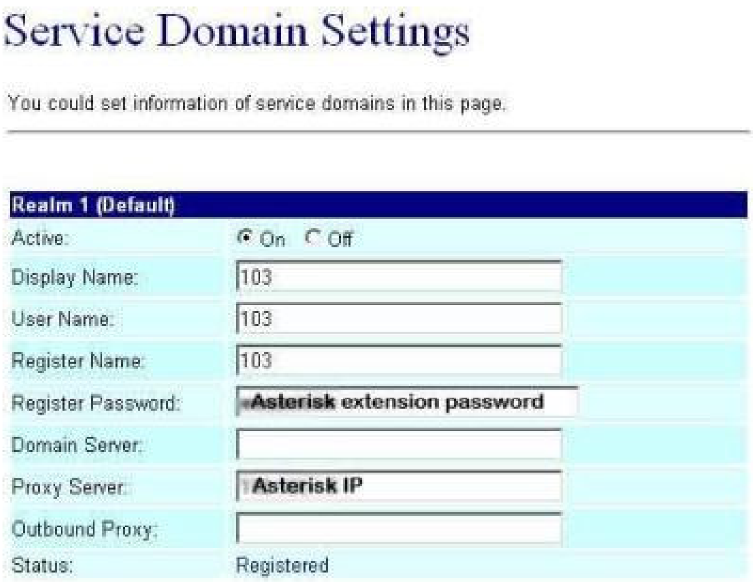
Here the '#' is important to avoid the two stage dialing when you give a call from Asterisk to GSM.

Mobile To LAN Table

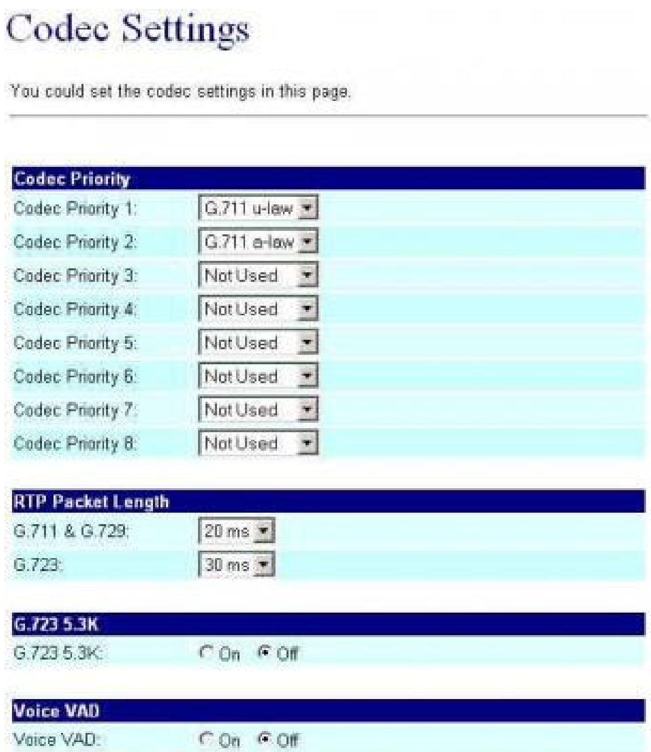
|@
Page: 1

Item	CID	URL	Select
0	authorised mobile n°	103	
1	another authorised n°	103	
2			
3			
4			
5			
6			
7			
8			
9			

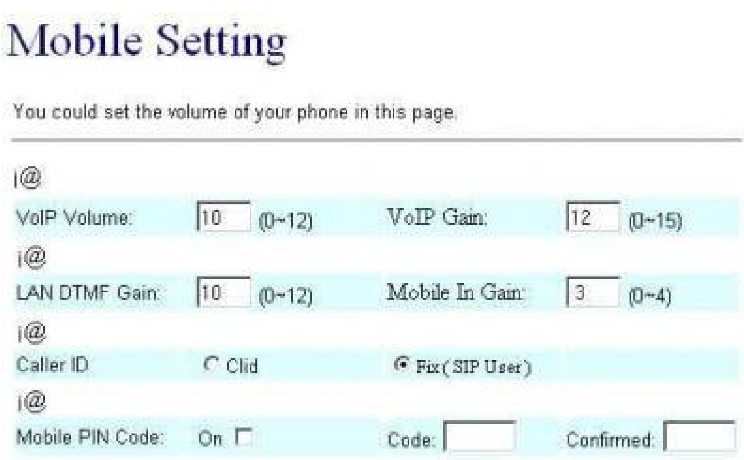
The mobile number you give in that page are the authorized mobile which can call GSM to Asterisk.
These mobile number must be defined as your GSM provider displays the number.
If you don't know how it is displayed, just give a call to the box and check the number given in the 'Incoming Mob' field of the 'Mobile Status' page.
Any number which is not in that list won't have acces to the LAN side, so to Asterisk. If you want to allow any number, just set '*' in that field ... but beware of the bill.



Once Asterisk configuration is made, you should get 'Registered' on the Realm1.



It is very important to use only ulaw or alaw when DTMF is Inband.
So if you want to be able to do some DISA when you call from GSM to Asterisk, it has to be one of these 2 codecs.



16.3 Antenna position

Another important thing is to properly place the provided antenna.
If your gsm reception is good, you should get around 18 or 19 as Signal Quality in the "Mobile Status" page.
With that level of signal quality, your audio quality will be very good.
Maximum signal quality = maximum audio quality.

16.4 Asterisk configuration

Once the GSM GATEWAY is set, you have to configure Asterisk.
On that side, you have to setup files as follow :

16.5 sip.conf

```
; GSM VOIP Gateway
[103]
type=friend
username=103
fromuser=103
regexten=103 ; When they register, create extension 401
secret=xxxxxxx ; Asterisk extension password
context=gateway ; Incoming calls context
dtmfmode=inband ; Very important for DISA to work
call-limit=1 ; Limit to 1 call max
callerid=GSM Gateway <103>
host=dynamic
nat=no ; Gateway is not behind a NAT router
canreinvite=no ; Typically set to NO if behind NAT
insecure=very
qualify=yes
disallow=all
allow=ulaw ; prefered codec for DTMF detection
allow=alaw
```

16.6 extensions.conf

```
; ******* GSM Gateway incoming calls **********
[gateway]
exten => _103,1,Answer()
exten => _103,2,DigitTimeout(3) ; give enough time to do second stage dialing
exten => _103,3,ResponseTimeout(5)
exten => _103,4,DISA(no-password|outgoing) ; here 'outgoing' is the normal context to deal
with the dial plan

[outgoing]
...
; example of LAN to GSM call
; call the GSM Gateway sim card mail box throug GSM
exten => _888,1,SetCallerID("xxxxxxxxxx")
exten => _888,2,Dial(SIP/${EXTEN}@103,60,r)
exten => _888,3,Hang
```